# COMPUTER-AIDED LABORATORY 1
# Signal and Spectrum Generation

## A. Discussion

### 1. Part 1, Signal Generation

In this part, the signals

$$s_1 = A_1 \cos(2\pi f_1 t),$$

$$s_2 = A_2 \cos(2\pi f_2 t), \ \&$$

$$s_3 = A_3 \cos(2\pi f_3 t)$$

are generated and will be plotted as *Plot 1* of *Figure 1*. You will be asked to enter values for the frequencies and amplitudes.

Before a signal can be plotted as a function of t, we have to establish a range for t. You will be asked to enter the minimum and maximum values for t.

After the individual cosine waves have been generated and plotted, the multitone signals

$$s_4 = s_1 + s_2, \ \&$$

$$s_5 = s_1 + s_2 + s_3$$

will be generated and plotted as *Plot 2* of *Figure* 1.

### 2. Part 2, Spectrum Generation

The Fourier transforms of $s_1$, $s_2$, and $s_3$ are computed and their magnitudes are plotted (in three different colors) as *Plot* 3 of *Figure* 2. The Fourier transform of $s_5$ is computed and plotted as *Plot* 4 in *Figure* 2.

### 3. Part 3, Flat Top Pulse

In this part you will be asked to enter an amplitude and pulse width for a flat top pulse. Don't use a pulse width in excess of 50% of the length of your time interval (final value - initial value) nor less than about 10% of the length of the interval.

The pulse waveform will be plotted as *Plot* 5 in *Figure* 3 (called $s_6$) and the magnitude of the Fourier transform of the pulse will be plotted as *Plot* 6 in *Figure* 3.

## B. PROCEDURE

### 1. Initial Values

Run the program **lab1** using the following values for your first run:

Part 1.
$t_{min} = 0.0$
$t_{max} = 0.1$
$f_1 = 40$, $A_1 = 5$
$f_2 = 80$, $A_2 = 2$
$f_3 = 150$, $A_3 = 1$

Part 2. none

Part 3. Pulse width, T = 0.02
Amplitude A = 10

### 2. User-Chosen Values

Run the program at least twice more using your own values for the parameters, observing the protocols discussed above. Record the values used and plot the figure windows.

### 3. Printing

Print the plots from Parts 1 and 2.

## C. REPORT

1. Compute the Fourier transforms for the signals $s_1$ through $s_6$ and sketch the transforms for the values you used in Part B. Clearly show all your equations and their evaluation for the parameters chosen.

2. Compare your theoretical calculations to the figures. Are there any discrepancies in the amplitudes of the frequency components or in the frequencies of the single and multitone signals or the zero crossings of the spectrum of the flat top pulse signal? Try to explain any discrepancies.

# COMPUTER-AIDED LABORATORY 2
# Fourier Transform Analysis

## A. Discussion

### 1. Part 1, Ideal Sampling Function

In this part, the signal

$$s_1 = \sum_{m=-\infty}^{\infty} \delta\,(t - mT_0),$$

is generated and will be plotted as the top plot of Figure 1 . This is seen to be an infinite string of delta functions or pulses. You will be asked to enter the separation between pulses, $T_0$.

The Fourier transform of $s_1$ will be computed and displayed in the bottom plot of that same figure.

### 2. Part 2, Rectangular Pulse

Now the signal will be defined as

$$s_2 = A\ rect\left(\frac{t}{T}\right).$$

You will enter A and T and the time domain signal will be plotted in the top plot of Figure 2.

The Fourier transform of $s_2$ will be computed and displayed in the bottom plot of Figure 2.

After we have seen the square pulse of $s_2$ and its transform, now we will delay the square pulse by half of a pulse width, so that

$$s_3 = A\ rect\left(\frac{t - T/2}{T}\right).$$

This signal and its transform will be plotted in the top and bottom, respectively, in Figure 3.

### 3. Part 3, Periodic Square Pulse Train

In Part 2 we saw the spectrum of a single square pulse. Now we want to let the square pulse be periodic with period $T_0$. You will specify the amplitude, pulse width T, and the period, $T_0$. Notice that if $T \geq T_0$ the input is dc. This signal and its spectrum are plotted in Figure 4.

4. Part 4, Time Domain Sinc Pulse

To utilize the duality property, we now let the time domain signal become a sinc pulse defined by

$$s_5 = A \ sinc\left(\frac{t}{T/2}\right),$$

where A is the amplitude at t = 0 and T is the time between the first nulls (around t = 0). You will enter A and T. The time domain signal and its transform will be plotted in Figure 5.

5. Part 5, Frequency Shifting

We now wish to observe the effects of frequency shifting. We will multiply the time domain signal of Part 4 by a cosine carrier wave of unity amplitude. This new signal can be written as

$$s_6 = A \ sinc\left(\frac{t}{T/2}\right) \cos\left(2\pi f_c t\right),$$

where $f_c$ is the carrier frequency. You will enter $f_c$. Figure 6 shows $s_6$ and its Fourier transform.

## B. PROCEDURE

1. Initial Values

Run the program **lab2** using the following values for your first run:

Part 1. $T_0$ = 500 mS

Part 2. A = 10 V
T = 100 mS

Part 3. Period, $T_0$ =500
Pulse width T =100
Amplitude A = 10

Part 4. Time between nulls T = 200 mS

Amplitude $A = 10$ V

Part 5. Carrier frequency $f_c = 100$ Hz

2. User-Chosen Values

Run the program at least twice more using your own values for the parameters, observing the protocols discussed above. Record the values used and plot the figure windows.

3. Printing

Print the plots from Parts 1 and 2.

## C. REPORT

1. Compute the Fourier transforms for the signals $s_1$ through $s_6$ and sketch the transforms for the values you used in Part B. Clearly show all your equations and their evaluation for the parameters chosen.

2. Compare your theoretical calculations to the figures. Are there any discrepancies in the amplitudes of the frequency components or in the frequencies? Try to explain any discrepancies.

3. In Part 1, why does the frequency spectrum consist of only delta functions?

4. We have learned that there is a one-to-one correspondence between a time domain function and its Fourier transform. In Part 2 we see that the magnitude of the Fourier transforms of $s_2$ and $s_3$ are identical even though $s_2$ and $s_3$ are not. How can this be reconciled.

5. How does the spectrum of Part 3 differ from the spectrum of Part 2? How are they the same? Explain why they are different and reconcile this with your answer for Question 3.

# COMPUTER-AIDED LABORATORY 3
# Ideal and Real Filtering in and out of Noise

## A. Discussion

### 1. Input Signal

For this entire lab, the input signal will be a square pulse with a duration of one millisecond. (Square pulses have wide use in communications: they are the standard pulse for digital communication whether it be within a computer, computer to computer, or in generating digital radio signals.) The amplitude of the pulse is nominally set to 1000 volts so that

$$s(t) = 1000 \ rect\left(\frac{t}{10^{-3}}\right).$$

The Fourier transform of the square pulse is a sinc pulse which extends from $-\infty < f < \infty$. The pulse and its transform are shown in Figure 1.

If we wish to transmit this pulse from one place to another, we must use a channel as a conduit of travel. This channel might be a transmission line, a fiber optic guide, or maybe an antenna transmitting to free space. Whatever the conduit, its bandwidth is limited. This means that even though the input signal has a frequency content over infinity, the channel will not pass all the frequencies of the input signal. Although this may be undesired, the channel filters the input signal so that the received signal, or the output, of the channel is not identical to the input.

One of the first questions that arises is that if the channel is limiting the frequencies of the input, how many of the original frequencies must pass for the output to be recognized as representing the input. In our case for this lab, this means can the output be recognized as resulting from a square pulse input. By definition, the measure of the frequencies passing through the channel (or filter) is the bandwidth. You will be allowed to experiment with different bandwidths to see how they affect the output signal.

To help us relate the results of this lab to square pulses of other amplitudes and duration, we will relate our filter bandwidths to the null bandwidth of the input signal (that is, the frequency of the first zero crossing). From theory, and as can be seen in the plot, the null bandwidth is 1000 Hz for our signal.

We will observe the response of real and ideal filters. You will probably discover some aspects of filtering you never thought of before. All our filters will have a maximum response of unity.

## 2. Part 1. Ideal Filter

We begin our filtering exercise with an ideal filter. An ideal filter is one which has a frequency response of one over its entire bandwidth. At the cutoff frequency, its response is vertical, from one to zero. The only variable of an ideal filter is its bandwidth. You will set the bandwidth, B, once for each running of this lab. Every filter will operate with the bandwidth that you specify. The frequency response of the ideal filter with the bandwidth that you enter is shown in Figure 2.

In Figure 3, you will see two plots. The top plot is the time domain output of the ideal filter with your bandwidth, with the input our square pulse. The bottom plot is the frequency domain output of the ideal filter.

## 3. Part 2. Real Filter

Now we want to use the power of simulation to approximate the response of a "real" filter. A real filter is one that we can build using electronic components, such as resistors, capacitors, and inductors. We will simulate a single-pole filter, such as you might find in a simple RC filter used for AM detection. The 3-dB bandwidth of the real filter is set to the same bandwidth, B, as you selected above. Figure 4 shows the frequency response of the real filter with the specified bandwidth.

In Figure 5, the output of the real filter with a square pulse input is shown. The top plot shows the time domain response and the bottom shows the response in the frequency domain. Notice that in the frequency domain the frequency response does not go to zero at the bandwidth limit.

## 4. Part 3. Frequency Discrimination

The purpose of a filter is to allow some frequencies to pass unattenuated while reducing or preventing other frequencies from reaching the output. Choosing some frequencies and rejecting others is a description of frequency discrimination, i.e., discriminating one frequency, or band of frequencies, from another. How well a filter can discriminate among the frequencies present is a function of the slope of the frequency response in the transition band of the filter. This in turn is controlled by the "Q" of the circuit: its quality factor.

To allow us to observe the discrimination qualities of our real and ideal filters, we multiply our input pulse by $2*\cos(2\pi f_c t)$, i.e., we will have a shifted input signal described by the equation

$$s_{shift}(t) = 2 \cdot 1000 \; rect\left(\frac{t}{10^{-3}}\right) \cos\,(2\pi f_c t).$$

We arbitrarily set the carrier frequency to 3 times the bandwidth, $f_c = 3B$.

From modulation or frequency shifting theory, we know that the Fourier transform of this product will be the original frequency spectrum repositioned at $\pm f_c$, each with half the original amplitude. Since the original frequency spectrum was an infinite duration sinc pulse, we will have two sinc pulses--one at $\pm f_c$. By using the scaling factor or 2 multiplied by the carrier, we position two sinc pulses at the original amplitude.

We will attempt to select the lower sideband of the modulated signal using real and ideal band-pass filters. We will place our ideal and real filters centered at $f_c$ - B/2. Our purpose is to try to recover the signal from the lower side band while rejecting all other frequency components. Figure 6 shows the modulated signal and the real and ideal LSB filters. In Figure 7 we see the result of the lower sideband filtering.

5. Part 4. Signal plus Noise.

We will now repeat the above experiments where the input signal will be the original square pulse with additive white gaussian noise added. Figure 8 shows the new input signal with the time domain on top and the frequency domain on bottom.

The ideal filtering of the signal plus noise using your bandwidth is shown in Figure 9, time and frequency domains on top and bottom respectively. Figure 10 shows the same for the real filter output. And, for the final plot, Figure 11 shows the time domain outputs of the real and ideal LSB filters.

## B. PROCEDURE

1. Initial Values

Run the program **lab3** using the following values for your first run:

Part 1. Bandwidth B = W (You can enter either W or 1000, your choice.)

2. User-Chosen Values

Run the program as many times as necessary using your own values for the bandwidth to answer the questions below. Record the values used and plot the figure windows.

3. Printing

Print the plots from Parts 1 and 2.

## C. REPORT

1. Determine the bandwidth required so that the outputs of the real and ideal filters are recognized as a square pulse. The required bandwidth will not be the same for the ideal and the real filters. (This <u>is</u> a subjective judgment.)

2. From Figure 4 in Part 2, how can you determine the bandwidth of this "real" filter frequency response?

3. Why is the time domain pulse of Figure 5 shifted to the right of the input pulse?

4. In Part 3 with B = W/2, which filter yields the better output? What about when B > 2W? Why?

5. In Part 4 with B > 4W, the outputs are not recognized as the inputs. Why?

6. Comparing Figures 3 and 5, we see that the "real" filter output is a better reproduction of the input than the "ideal" filter. With this observation it would seem that the real is superior to the ideal. How do you resolve this paradox?

# COMPUTER-AIDED LABORATORY 4
# Sampling and Recovery

## A. Discussion

Part 1: Signal generation and sampling

In this part, the input signal

$$s = 2[\cos(2\pi(0.25 f_3)t) + \cos(2\pi(0.5 f_3)t) + \cos(2\pi f_3 t)]$$

is generated. You will be asked to enter $f_3$, the highest frequency in the input analog signal. You may choose any value between 100 Hz and 1000 Hz. You will then be asked to enter $f_0$, the sampling frequency. The signal s and an ideal sampling waveform (the computer's best approximation to a delta function sequence) will be plotted as Plot 1 of Figure 1. The ideal sampled waveform will be plotted as Plot 2 of Figure 1.

You are asked to enter the duty cycle, in percent, for the pulses of a flattop pulse sampling function. The duty cycle in percent is defined as

$$d = \frac{T}{T_0} \ x \ 100$$

where T is the width of the flattop pulses and $T_0$ is the time between pulses ($T_0 = 1/f_0$). You should use values for d between 10 and 90 percent. If too small a value is entered for d, you will be prompted to enter a larger value. The flattop sampled signal will also be plotted in Plot 2 of Figure 1 along with the ideal sampled signal.

Plot 3 of Figure 2 is a plot of the spectrum of s. Plot 4 of Figure 2 is a plot of the spectrum of the ideal sampled signal si. Plot 5 of Figure 3 is the spectrum of the flattop sampled signal sf. Plot 6 of Figure 3 is the spectrum of a single flattop pulse of the width you used for flattop sampling. For these plots you will be asked to enter the maximum of the frequency scale. You should use something between 2 and 4 times the sampling frequency you choose, but not greater than 5000 Hz.

Part 2- Signal recovery

In this part you will attempt to recover the original signal s from the sampled signals using an ideal low pass filter. You will be asked to enter the cutoff frequency of the ideal low pass filter. Plot 7 of Figure 4 will show the spectrum of the ideal sampled signal coming out of the ideal low pass filter. Plot 8 of Figure 4 is the actual output signal from the filter in the time domain. It is called "sri" for recovered signal, ideal. The error signal e=s-sri is shown in red in Plot 8.

Plot 9 of Figure 5 shows the spectrum of the flattop sampled signal coming out of the ideal low pass filter. Note that we have made no correction for the dropoff of the high frequency components. Plot 10 is the actual output signal from the filter in the time domain. It is called "srf" for recovered signal, flattop. The error signal e=s-srf is shown in red in Plot 10.

**B. Procedure**

1. Initial Values

a. Run the program **lab4** using the following values for the parameters;

Highest freq.: f3 = 400 Hz
Sampling freq.: fo = 1000 samples per sec.
Duty cycle: . d = 80 percent
Max of Freq. plots: Fmax =4000 Hz
LPF cutoff: fc = 500 Hz.

b. Plot the Figure windows using the print command.

2. User-Chosen Values

a. Run the program several more times trying different combinations of the parameters. For some multiples of highest freq. and sampling freq., the plots may smear out a bit and some of the amplitudes will vary. This is due to digital Fourier transform techniques used in the computer.

b. Collect two more sets of plots. One of these should be with the same parameters as used above except with a smaller duty cycle in the flattop pulse sampling function. Observe the reduced error in the final plot. The other should be an undersampled case, where an insufficient sampling frequency is used. Observe what happens to the error in both Plots 8 and 10.

**C. Report**

1. Compute and sketch the Fourier transforms for the signal s, for the ideal sampled signal si and for the flattop sampled signal sf for the three sets of plots you have collected. Compare these to your experimental results and comment.

2. Discuss your choice for the cutoff frequency you used in the low pass filter for signal recovery. Comment on the effect of duty cycle (pulse width) on error in the recovered signal in the last plot.

# COMPUTER-AIDED LABORATORY 5
## Quantization and Companding

## A. Discussion

Part 1: Signal generation, uniform quantization and sampling

In this part, the input signal

$$s = \cos\,(5000\pi t)[sinc\,(SWt) \;**\; comb(100t/3)] + 0.1\cos(1000\pi t)$$

is generated. The ** operation represents convolution and the comb function represents a sequence of impulses separated by an interval of 0.03. You will be asked to enter SW , the highest frequency of the sinc pulses. You may choose any value between 100 Hz and 2500 Hz. You are then asked to enter the number of quantization levels. The quantizer in this laboratory is a midtread quantizer, so an odd number must be input.

Plot 1 is a display of the message signal. Plot 2 is a uniformly quantized version of the message signal. Plot 3 is the power spectral density of the message signal, and the signal power is computed based on the provided sinc bandwidth.

You are asked to enter a value for mu to determine the mu-law compression characteristic defined by:

$$v_{out} = \frac{\log\,(1 + \mu|v_{in}|)}{\log\,(1 + \mu)}$$

This characteristic is shown in plot 4 for the value of mu input. The characteristic is then applied to the message signal and the resulting compressed signal is illustrated in plot 5.

The compressed message signal is quantized using the number of levels supplied above and displayed in plot 6. This quantized signal is then expanded using the inverse of the above transfer characteristic and shown in plot 7.

Plots 8 and 9 illustrate different perspectives of the error signal associated with uniform quantization (shown in green) and non-uniform quantization via companding (shown in red).

## B. Procedure

### 1. Initial Values

a. Run the program **lab5** using the following values for the parameters;

| | |
|---|---|
| Sinc bandwidth: | SW = 2000 Hz |
| Number of quantization levels: | L = 9 |
| Compressor characteristic constant: | mu = 10 |

b. Plot the Figure windows using the print command.

### 2. User-Chosen Values

a. Run the program several more times trying different combinations of the parameters.

b. Collect three more sets of plots. One of these should be with the same parameters as used above except with a much higher value for mu (> 2000). Another should use a larger number of quantization levels, to demonstrate where non-uniform quantizing becomes unnecessary. A third should use a smaller sinc bandwidth. Observe what happens with the power and quantization error in this case.

## C. Report

1. Compute the signal to uniform quantization noise ratio (SNR) for the signal parameters of each plot. Compute signal to non-uniform quantization noise ratio using SNR = 6n + 4.8 - 20log[ln(1 + mu)] dB, where n is number of bits of quantizer codeword and is equal to log2 L. Show how you calculated values signal power and uniform quantization noise power. Comment on how your values compare with the computed value.

2. Prepare a plot of uniform SNR (in decibels) and non-uniform SNR (in decibels) vs. signal power (also in decibels) for various values of mu and SW. Try to determine at what signal power the two curves intersect.

3. Discuss the effect of increasing the number of quantization levels.

4. Comment on where one quantization method outperforms the other. Discuss why this occurs.

# COMPUTER-AIDED LABORATORY 6
# Amplitude Modulation

## A. Discussion

Part 1: AM modulation with a sinusoid test tone

In this part, ordinary AM and DSBSC AM signals are generated using the test sinusoid message signal

$$s = \cos(2\pi f_m t).$$

You will be asked to enter $f_m$, the frequency of the sinusoid. You may choose any value between 50 and 400 Hz. You will then be asked to enter μ, the modulation index and $A_c$, the carrier frequency amplitude.

The signal

$$k_{am} = \mu \cdot m$$

is used to generate the ordinary AM signal

$$s_1 = A_c\ (1 + k_{am})\ \cos(2\pi f_c t).$$

The carrier frequency $f_c$ is set to 2000 Hz in the program.

The sinusoid message signal m and $1 + k_{am}$ are plotted as Plot 1 of Figure 1. The ordinary AM signal $s_1$ is plotted as Plot 2 of Figure 1.

Plot 3 of Figure 2 is a plot of the spectrum of the message m. Plot 4 of Figure 2 is a plot of the spectrum of the AM signal $s_1$.

The DSBSC signal

$$s_2 = A_c\ m\ \cos(2\pi f_c t)$$

is generated and plotted as Plot 5 of Figure 3. The spectrum of $s_2$ is Plot 6 of Figure 3.

Part 2: The AM modulation and demodulation process with a multi-tone message signal.

In this part you will use the message signal

$$m = \cos(2\pi f_m t) + \cos(2\pi(0.5 f_m)t) + \sin(2\pi(0.25 f_m)t)$$

to create ordinary AM signal $s_1$ with a modulation index μ, DSBSC signal $s_2$, and SSBSC signal $s_3$. The value of the AM gain constant $k_a$ for the ordinary AM signal is computed as

$$k_a = \frac{\mu}{\max(abs(m))}$$

and is printed out for you. Thus the signal $k_{am} = k_a m$ will have a maximum absolute value of μ as required.

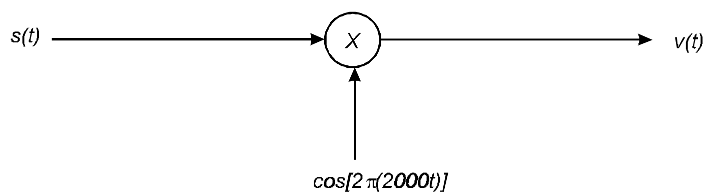
Plot 7 of Figure 4 is a plot of m and $1 + k_{am}$. Plot 8 of Figure 4 is the ordinary AM signal $s_1$.

Plot 9 of Figure 5 is the spectrum of multi-tone message m. Plot 10 is the spectrum of the AM signal $s_1$.

Next, we recover the message signal m from the ordinary AM signal $s_1$ using an envelope detector. Plot 11 of Figure 6 shows the envelope signal with the bias removed and the gain restored (dividing by $k_a A_c$). The original message signal m is shown for comparison.

Plot 13 of Figure 7 shows the DSBSC signal $s_2$ using the multi-tone message signal m. Plot 14 of Figure 7 shows the spectrum of $s_2$.

Plot 15 of Figure 8 shows the SSBSC signal $s_3$ obtained by passing $s_2$ through a SSB filter. Plot 16 of Figure 8 shows the spectrum of $s_3$.




Referring to the diagram above of a coherent detector, Plot 17 of Figure 9 shows spectrum x, the output of the multiplier when the input is $s_2$, the DSBSC signal. Plot 18 of Figure 9 shows the spectrum of x when the input is $s_3$, the SSBSC signal.

Plot 19 of Figure 10 shows $v_2$, the output of the coherent detector, in the time domain, for the DSBSC signal. Plot 20 of Figure 10 shows $v_3$, the output of the coherent detector for the SSBSC signal. You will be asked to enter $f_{co}$, the cutoff frequency for the LPF.

## B. Procedure

### 1. Initial Values

a. Run the program **lab6** using the following values for the parameters;

| | |
|---|---|
| Freq. of message: | $f_3$ = 200 Hz |
| AM modulation index: | $\mu$ = .8 |
| Carrier Amplitude: | $A_c$ = 4 |
| LPF cutoff: | $f_{co}$ = 500 Hz |

b. Plot the Figure windows using the print command.

### 2. User-Chosen Values

a. Run the program several more times trying different combinations of the parameters.

b. Collect only one more sets of plots. This set should use a value for the modulation index $\mu$ that is greater than one to create a situation of overmodulation for the ordinary AM. Observe what happens to the output of the envelope detector for this case.

## C. Report

1. Compute and sketch the Fourier transforms for the message signal m and for the three signals $s_1$, $s_2$, and $s_3$ for both the single tone and multi-tone message signals. Compute and sketch the Fourier transform for the multiplier outputs $x_2$ and $x_3$ for DSBSC input $s_2$ and the SSBSC input $s_3$. Compare these to your experimental results and comment.

2. Is this SSB transmission using the upper or lower sideband?

3. Discuss your choice for the cutoff frequency you used in the low pass filter for signal recovery.

# COMPUTER-AIDED LABORATORY 7
## Signal Demodulation

### A. Discussion

Lab 7 is organized completely different than the preceding labs. In this lab, all interaction between the user and the program will be via graphical user interfaces (GUIs). The only time you will type on the command line is to enter "lab7".

To allow the various parts of the GUI to operate, you must take some initial action. After activating MATLAB and locating yourself in the EO3513 directory, position the cursor over "Options". Click the left mouse button, and the pull down menu will show several choices. Place the cursor on the third item. If it says "Disable Background Processes" it is OK and you can put the cursor within the command window somewhere and click. If, however, it says "Enable Background Processes", you must click on it.

1. Part 1.

a. Modulating Signal

In this part, the modulating signal is a single-frequency sinusoid

$$m(t) = \cos\ (2\pi f_m t).$$

You will specify the frequency of m(t), $f_m$. The time and frequency domain representations of m(t) are plotted Figure 1. You will be given an opportunity to print the plot, then it will be cleared from the screen. The carrier frequency will be computed as ten times $f_m$,

$$f_c = 10\ f_m.$$

Both the modulating and carrier waves will have an amplitude of one.

b. Menu Selection

After the modulating signal has been specified, a menu will appear on the screen. Upon that menu you have several choices which are: Envelope Detector

Coherent Detector

Phase-Locked Loop

Costas Loop, and

Continue.

By selecting one of the items (by clicking on it), you will observe the detector at its various stages. For the envelope detector the modulated signal is AM with carrier. For the coherent detector and Costas Loop, the signal is DSB suppressed carrier. And, for the PLL, the modulated signal is FM. The continue button takes you to Part 2 of the lab.

There are various other informational buttons and windows on the screen as well. Feel free to look at them. If you select "Close", you terminate the lab and you will have to restart.

You will be asked for additional information when you select the demos. In the envelope detector, you will specify the modulation index of the AM modulation. For the coherent detector and the costas loop, you specify the phase difference between the carrier of the incoming modulated signal and that of the local oscillator. In the PLL you will asked to select the beta for the FM modulation.

You may run each and every demo as many times as you wish. Each demo will use the modulating signal that you selected in the beginning. Plots will be displayed of the various signals, and you will be given the opportunity to print them, after which they will be cleared. When you are satisfied with the demos, select "Continue" to proceed to Part 2.

2. Part 2

After selecting "Continue" you will see a second menu. Besides the informational buttons and windows from before, you will see two columns. The one on the left under "Signals" is for the type of modulation that you wish to generate. Once you have made a selection, you will need to specify the same information as above. You will also need the number of modulating frequencies, and what those frequencies are. You will be asked for the amplitude of the modulating frequencies. No plots will be generated in the modulation part.

After you have generated a modulated signal, you then must make a selection under "Detectors". You will not be asked for the phase difference for the coherent detector and costas loop as we have arbitrarily set them at 0 and $\pi/4$ respectively for this part of the lab. For any of the detectors, the object of demodulation is the lowest of the frequencies that you selected for the modulating signal, that is, the cutoff frequency for the LPF is set at the lowest modulating frequency. The carrier frequency will be computed as 10 times the lowest modulating frequency.

You may choose to demodulate your signal with any and all of the detectors. For example, you can create AM with carrier and see if the coherent detector will demodulate it. Plots of the (attempted) demodulated signal will be displayed. You will be given an opportunity to print them, after which they will be cleared.

You may run the demos as many times as you wish. Click on "Close" or "Quit" when you are finished.

## B. PROCEDURE

1. Initial Values

Run the program **lab7** using the following values for your first run:

Part 1. $f_m$ = 500

Part 2. No. frequencies = 3
frequencies = default
Amplitude = 1
$\mu$ = 1 (AM with carrier)
$\beta$ = 8 (FM)

2. User-Chosen Values

Run the demos as many times as necessary to answer the questions below.

3. Printing

Print the plots from Parts 1 and 2.

## C. REPORT

1. In Part 1, compare the time and frequency domain outputs of the different modulator stages with theory. Comment on the results.

2. Narrow band FM is created when $\beta \leq 0.2$. Use the PLL demo to determine if choosing values above and below this limit conforms to theory.

3. In Part 2, try every demodulator on every modulation type. Explain, with necessary equations, whether or not the demodulation was successful.

4. The low-pass filters of the demodulators use a "real" second order filter. To see the implications of this, first choose your favorite signal-detector pair. Select the signal, and place the first and second frequency only 100 Hz apart, e.g., Freq1 = 400 and Freq2 =500, then demodulate. Now, do the same thing again, but select the first and second frequency to be at least 300 Hz apart. Comment on your observations.